



SayHi™

ES330 IP PHONE

User Manual



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www.escene.hk 1

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1.Getting Started

About

ES330 is a small-screen-based IP phone in Sayhi IP phone series, it has fashion and technological appearance, excellent voice quality, and powerful features, and it is a new generation of intelligent phones to replace of the traditional desktop office terminals, It accomplished the powerful telephony features by cooperating with the communications platform,. such as the call transfer, hotline, three-party conference calling, speed dial, voice mail, Do Not Disturb, etc.

Feature Highlights:

- 128*64 Pixel LCD with Support Chinese display
- HD Voice: HD Codec
- 2 VoIP accounts
- Enterprise Phone Book
- 12 programmable hard keys and support BLF
- Expansion Module: Up to 6 EMS32 programmable key modules
- Support Plug and Play
- Support PoE and AC power adapter

Technical Features

Item	ES330
Screen	Grayscale LCD with background light
	128*64 pixel
Line	3
Function Keys	4 Soft keys, 2 Line keys (dual-color LED) 6 Navigation keys (arrow button, OK button, C button) Volume adjust, Hands-free, Mute, Headset, Message, Menu, Directory, Service, Hold, Redial, Conference, Transfer
VoIP Protocol	SIP 2.0
Network Protocol	HTTP, BOOTP, TFTP, *IEEE 802.1Q, *IEEE 802.1X
Codec	G.723.1, G.729 A, G.711 A/U, G.722
QoS	TOS, Jitter Buffer, VAD, CNG, G.168 (32ms)

Network	2*RJ45 10/100M Ethernet interfaces (LAN/PC) IP Assignment: Static IP or DHCP VPN(L2TP) DNS Clients (Primary and Secondary)
Conversation Mode	Handset, Headset or Hand-free Mode, Call center headset and 3.5mm headset supported 9-levels volume adjustment
Call Processing	Call Sharing/Bridged Lines Line Status Indicator(dual color LED) Multi Account Call Waiting, Call Queuing, Line Switching Call Forward, Call Transfer, Call Holding, Call Pickup, Callback One Key Dial, Redial Phone directory speed dial, Call record direct dial 3-way conference DnD Voice mail, Voice Prompt, Voice Message BLF
Expansion Interface	EXT: ESM32 programmable key module SOR: FXS, FXO, Record USB
Security	User Authentication for configuration pages Signaling encryption Media encryption
Application	*Public phone directory Private phone directory *PnP (Plug & Play)
Power Supply	Power adapter: AC 100-240V input and DC 12V/1A output PoE (IEEE 802.af)
Specification	Storage Temperature: 0 -60 Operating Humidity: 10%-90% Size 254mm*205mm*87mm

2.Connecting Your Phone

Your system administrator will likely connect your new SayHi ES330 IP Phone to the corporate IP telephony network. If that is not the case, refer to the graphic and table below to connect your phone.

- 1) Open the box ES330 IP Phone; carefully check the packing list, Packing List as follows:

Item	Counts
IP Phone	1
Handset	1
Handset Cord	1
Power adapter	1
RJ45 cable	1
CD	1
Quick Installation	1
Quick User Guide	1
Product certification	1

2) As shown in figure 2.1 and figure 2.2, Please plug Handset Cord into RJ11 interface(IP Phone and Handset), RJ45 cable into the LAN interface ; IP Phone will automatically start if IP Phone with POE function.

3) The phone must work together with power adapter without POE support.

4) Connect your computer to PC interface of the phone with cable.

* More detailed description please refers to the *3.Phone overview-Understanding phone buttons and hardware*.

Figure 2.1 Interfaces of SayHi ES330

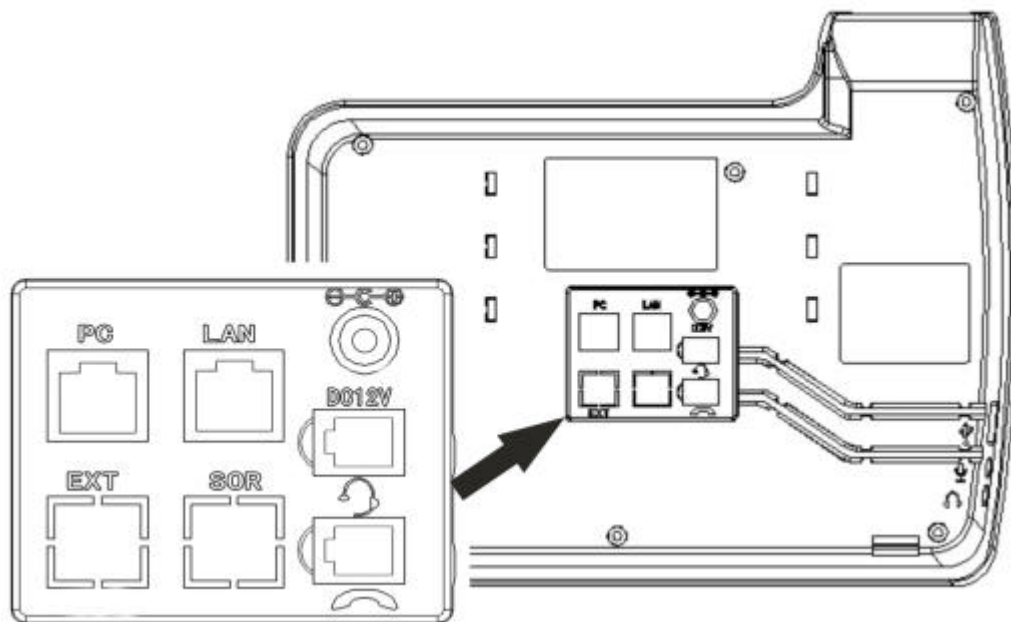
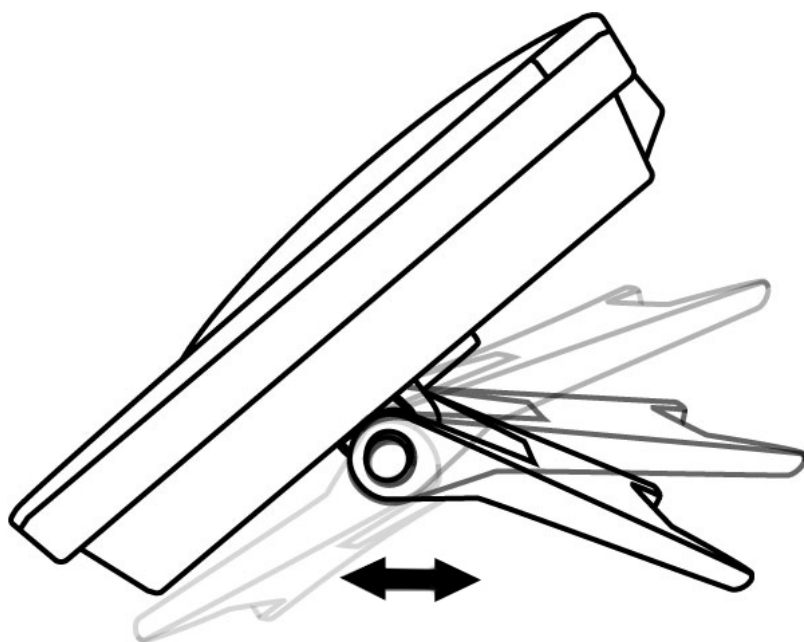


Figure 2.2 Footstand of ES330



5) If you want to connect a ESM (Expansion Module), you could connect it (any interface) to the EXT interface on the phone with a RJ-45 cable (a straight-through cable), and the second ESM can be connected to the first one too. It supports 6 ESMs.







3.Phone overview









Understanding Buttons and Hardware

From figure 3.1 to figure 3.2, you can understand buttons and hardware about SayHi ES330.

Figure 3.1 Buttons and Hardware of SayHi ES330



Num	Buttons	Description
1		Headset button: Toggles the headset on or off.  Red means the feature is enabled.
2		Mute button: Toggles the Mute feature on or off.  Red means the feature is enabled.
3		Messages button: Typically auto-dials your voice message service.  Red means have unread voice mail.

4	SERVICE	Server button: Open or Close the Services menu.
5	DIRECTORIES	Directory button: Use it to access call logs and corporate directories.
6	MENU	Menu button: Allows you to scroll through menus.
7		Volume button: Controls the volume and other settings.
8	CONFERENCE	Conference button: Connect calling / called party to the conference
9	REDIAL	Redial button: To Redial the last number.
10	TRANSFER	Transfer button: Transfer redirects a connected.
11	HOLD	Hold button: Put a call on hold
12	0-9, *, #	Basic Call Handling: press “#” send out a call(default)
13	Speaker button	Speaker button: Toggles the speakerphone on or off.  Red, steady: Pick up and enter normal call.
14	Softkey	Each displays a softkey function, To activate a softkey, press the softkey button.
15	Line buttons	Select the phone line (Call or Answer) ; Different colors for different status: 1)  Red, flashing: There is an incoming call. 2)  Red, steady: Pick up and enter normal call. 3)  Yellow-green, flashing: Holding call. 4)  Yellow-green, steady: Active call.
16	Programmable Buttons	Hotline number can be used to bind in order to achieve speed dial ; Turn on BLF: 1)  Red, steady: Remote line is busying. 2)  Yellow-green, steady: Remote line is idle. The order of the hot keys : On the left top to bottom: 1, 2, 3, 4, 5, 6 ; On the right top to bottom: 7, 8, 9, 10, 11, 12;



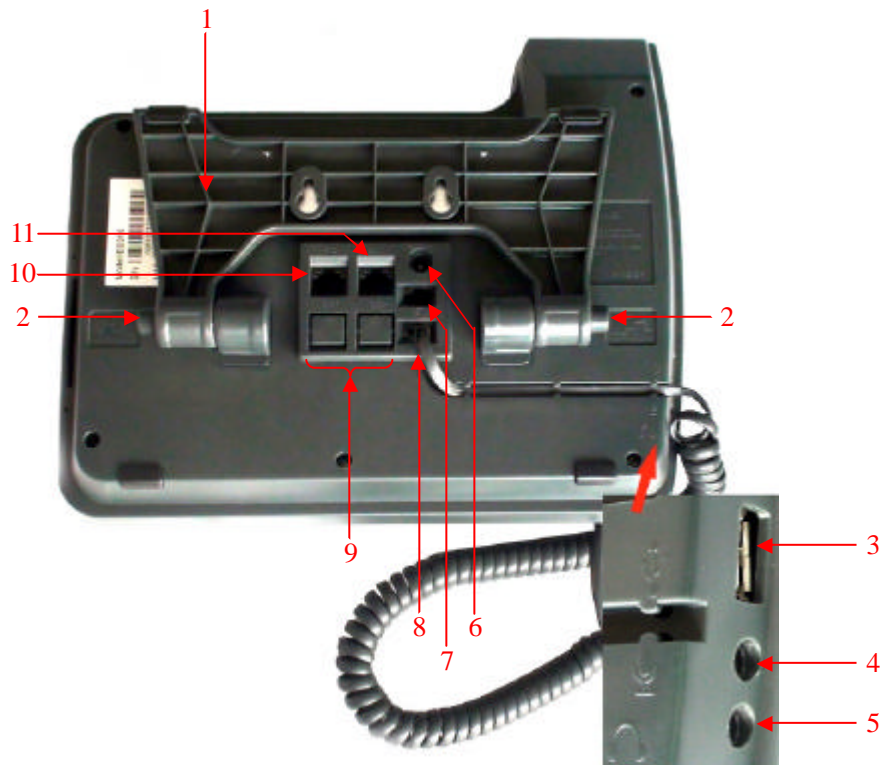
17	C	Back button: Return to the standby interface;
18	Navigation button	“Up”: Adjust ring volume, operate with the “down” button “Down”: Open ‘Missed Calls’ list; “Left”: Open “Received Calls” list; “Right”: Open “Dialed Numbers” list
19	OK	OK button: To confirm the action;
20	Hands-free speakerphone	Hands-free voice of the output
21	LCD screen	160*32 pixels, grayscale LCD with background light.
22	Light strip	 Red flashing: There are incoming call ;  Red, steady: Missed Calls, or phone busy;
23	Hands-free microphone	Sounds input when hands-free

Figure 3.2 Interfaces of SayHi ES330



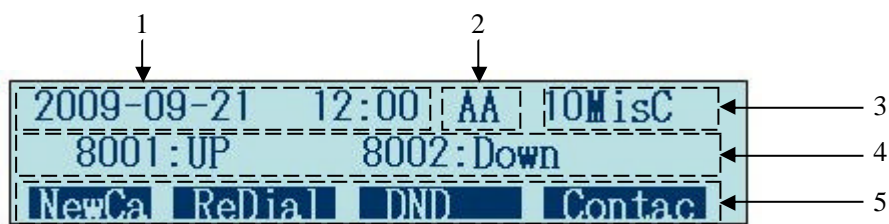
Num	Hardware	functions
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1	Footstand	Hold up phone
2	Footstand button	Press buttons at the same time to adjust the angle
3	Reserved for USB port	Enhanced scalability
4	Microphone port	Connect the Microphone
5	Headphone port	Connect the Headphone
6	Power port	12VDC
7	Headset port	Support RJ11 interface connection
8	Handset port	Connect the Handset
9	Reserved port	Enhanced scalability: 1) EXT: ESM interface ; 2) SOR: S-FXS O-FXO R-record
10	LAN port	Connect to a LAN interconnecting device
11	PC port	Connect to a local PC

Understanding Phone Screen Features

This is what your main phone screen might look like:

Figure 3.3 LCD of SayHi ES330



Num	Screen	Functions
1	Time and Date	Show current time and date.
2	Auto-answer	Enabled Auto-answer, displays "AA"
3	Missed calls	Show the number of missed calls.

4	Line status	<p>Show the phone line status:</p> <p>1) LAN: Disconnect : Disconnect into network.</p> <p>2) Peer-to-Peer : Only Peer-to-Peer call.</p> <p>3) 8001:Down : Network connected normal, but the line is not successfully registered.</p> <p>4) 8001:UP : Network is OK and the line is available.</p> <p>5) 8001:DND : Line is turned on DND.</p>
5	Softkey labels	Each displays a softkey function (displayed on your phone screen), and the function is different when menu changes.


4.Basic Call Handling

You can perform basic call-handling tasks using a range of features and services. Feature availability can vary; see your system administrator for more information.

Note: The bold type of the following text and following a “button” in table signifies the phone's button (for example, **OK** button), and the **NewCa** signifies softkey.

Placing a Call

Here are some easy ways to place a call on SayHi ES330 IP Phone:

If you want to...	Then...	
Place a call using the handset	Pick up the handset	--1) You can hear the dial tone ; --2) The first line light is  ; --3) Enter a number ; --4) Press '#' button (default) , -or press Send ; -or wait 5s (default), then it send the number automatically.
Place a call using a speakerphone	Press Speaker , or Programmable buttons or NewCa	
Place a call using a headset	Put on your headset and active Headset button, and then do as using speakerphone	





Redial	<p>--Press REDIAL button to dial the last number</p> <p>-or press Navigation button-Right > “Dialed number”, select a number, and press Dial or OK button.</p>
Dial from a call log	<p>--1) Press MENU or OK button > “Call history”, you can select “Missed calls”, “Received calls” and “Dialed numbers”,</p> <p>- or press Navigation button (in Standby interface) > select “Missed calls” (down), “Received calls” (left) and “Dialed numbers” (right) ;</p> <p>--2) Then press OK button or Dial.</p>
Place a call while Another call is active	<p>--1) Press Hold button or Hold;</p> <p>--2) Enter a number;</p> <p>--3) Press ‘#’ button (default) ;</p> <p>-or press Send to send the number.</p>




Tips

- You can dial on-hook, without a dial tone (pre-dial). To pre-dial, enter a number, and then go off-hook by lifting the handset or pressing **Send**, **Headset** or **Speaker** button.
- If you make a mistake while dialing, press **C** button to erase digits.

Answering a Call






You can answer a call by simply lifting the handset, or you can use other options if they are available on SayHi ES330.

If you want to...	Then...	
Answer with a handset	<p>--1) Your phone ring;</p> <p>--2) Line button of the ringing line is Red  and flashing, Light strip is Red  and flashing ;</p>	--Pick up the handset
Answer with the speakerphone (Non-headset mode)		<p>--Press Speaker button</p> <p>-or press the flashing  Line button,</p> <p>-or press Ans</p>
Answer with the a headset		<p>--Put on headset, press Headset button so that the status light is Red , and then do as using speakerphone</p>

Switch from a connected Call to answer a ringing call	--1) Another Line button is Red  and flashing, Light strip is Red  and flashing; --2) Press the flashing  Line button to answer (at this time, the original call will be hold.)
Auto-answer	--1) Press MENU or OK button > “Function setting” > “Auto answer”; --2) Select “Enable”; --3) Your phone answers incoming calls automatically after a few rings.

Ending a Call

To end a call, hang up. Here are some more details.

If you want to...	Then...
Hang up while using the Handset	--Return the handset to its cradle, -or press 
Hang up while using the Speakerphone	--Press Speaker button that is Red  , -or press Line button for the appropriate line, -or press 
Hang up while using the Headset	--Press Handset button, (Do not keep the headset mode) , -or press  (keep the headset mode)
Hang up one call, but preserve another call on the other line	--Press  , -or refer to the above three methods


Using Hold and Resume (Switch Calling Line)

You can hold and resume calls. You can take a call in one line at anytime, and the other lines would be hold. As a result of that, you can switch different calling line on our phone.

If you want to...	Then...
-------------------	---------

Put a call on hold	--Press HOLD button, -or press Hold
Hold a line and switch to another line	Press another Line button for the appropriate line
Resume a call on current line	--Press HOLD button, -or press Hold
Release a call on different line	Select the line want to release hold, press the line, so recovery;

Tips

- Engaging the Hold feature typically generates music or a beeping tone.
- A held call is indicated by the Yellow-green  and flashing Line button.

Transferring Calls


Transfer redirects a connected call. The target is the number to which you want to transfer the call.

If you want to...	Then...
Talk to the transfer recipient before transferring a call (consult transfer)	--1) Press TRANSFER button or press Transf ; --2) Enter number; --3) press “#” (default) , -or press Send then transfer the call, -or wait five seconds(default)then transfer the call
Transferred to idle lines or other numbers without talking to the transfer recipient (Blind transfer)	--1) Press TRANSFER button or Transf ; --2) Press Blind ; --3) Enter number; --4) Press “#” (default) -or press Send , then transfer the call; -or wait five seconds(default)then transfer the call

Blind transfer to the held line	--1) Press TRANSFER button or press Transf ; --2) Press the Line button of held line
---------------------------------	--

Using Mute

With Mute enabled, you can hear other parties on a call but they cannot hear you. You can use mute in conjunction with the handset, speakerphone, or a headset.

If you want to...	Then...
Toggle Mute on	Press Mute button, then the button is Red 
Toggle Mute off	Press Mute button, then the button light off

Do Not Disturb

You can use the Do Not Disturb(DND) feature to block incoming calls on your phone with a busy tone (Can also be set to their voice mail or other extension numbers, etc.).

If you want to...	Then...
Enable global DND	--1) Press DND ; --2) All enabled line on the phone would changes to 8001:DND status.
Enable DND on a single line	Press MENU or OK button > “Function setting” > “DND” > (select line) “Enable”
Disable DND	--Global DND enabled, press DND to disable global DND; --Line DND enabled, press twice DND , -or press MENU or OK button > “Function setting” > “DND” >(select line) “Disable”

3-way Conference

You can establish a three-party conference, during the conversation three phone parties can communicate with each other.

If you want to...	Then...
Invite the transfer recipient into a conference in a transferring	<p>--1) When the transfer recipient answer the call, press CONFERENCE button or Confe on your phone;</p> <p>--2) Then the held one, transfer recipient and you will be into a conference, and the LCD will display conferenc 0:0:10 status.</p>
Invite the third party into a conference in a active call	<p>--1) Press CONFERENCE button or Confe in an active call;</p> <p>--2) Enter the third party number ;</p> <p>--3) After connected the third party, press CONFERENCE button or Confe again</p>
establish a conference with held line	<p>--1) when one phone line is holding on and the other line is busy;</p> <p>--2) Press CONFERENCE button,</p> <p>-or Press Confe Soft key</p> <p>--3) press the held line's programmable button, the 3-way Conference will establish.</p>

5.Advanced Call Handling

Speed Dialing

Speed dialing allows you to enter an index number, press a button, or select a phone screen item to place a call.

If you want to...	Then...
Set up Speed Dials on your phone	--1) Press MENU or OK button > “Function setting” > “Hot line keys”; --2) You can configure twelve speed dial numbers on the SayHi ES330 IP Phone; --3) Press OK button or Modif to set and modify: -Mode: -Speed dial: Speed dial mode -Asterisk BLF: In the Speed dial based on the increase in BLF (Busy line detection) function -Account: Speed Dial hot keys using the account -Name: Description of this hot-key, -Number: Need to speed dial numbers --4) Press Submit to save the changes


Using the phone book

You can store a large number of contacts in your phone's directory. You can add, edit, delete, dial, or search for a contact in this directory.

If you want to...	Then...
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Add Contacts	<p>--1) Press Contac,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Press Modify;</p> <p>--3) Select “Add contact”, press OK button or OK;</p> <p>--4) Use the navigation keys to select content, press OK button or Modify to set and modify:</p> <p>-Name: set the name of contact,</p> <p>-NO.1-5: you can set up 5 contacts’ numbers,</p> <p>-Group: the contacts be divided into different user’s groups</p> <p>--5) Press Submit soft key to complete</p>
Add group	<p>--1) Press Contac soft key,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Press Modify soft key;</p> <p>--3) Select the “add group” then press OK button or OK;</p> <p>--4) Use the navigation keys to select content, press OK button or Modify to set and modify:</p> <p>-Group name: name of the group</p> <p>-Description: description of the group</p> <p>--5) Press Submit soft key to complete</p>

Modify group	<p>--1) Press Contac soft key,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Press Modify soft key;</p> <p>--3) Select the “Modify group” then press OK button or press OK ;</p> <p>--4) Select the group you want to modify, press the OK button or Modif to set and modify, press Submit to save the change</p>
Delete group	<p>--1) Press Contac soft key,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Press Modify soft key ;</p> <p>--3) Select the “Delete group” or OK button or OK ;</p> <p>--4) Select a group you want to delete, press OK button or OK</p>
View/Edit Contacts	<p>--1) Press Contac soft key,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Select “View ALL”,</p> <p>-or select a contact who are belong to different group;</p> <p>--3) Select the contact, press the OK button or View (to edit the contact’s information, press OK button or Modif)</p>

Call from phone book	<p>--1) Press Contact soft key,</p> <p>-or press MENU button > “Phone book”,</p> <p>-or press Directories button > “Phone book”;</p> <p>--2) Select “View ALL”,</p> <p>-or select a contact who are belong to different group;</p> <p>--3) Select a contact, then press Dial,</p> <p>(If there are multiple numbers of one contact, press Dial to enter the interface of “call options”, select the one you want to call and press Dial)</p>
Modify the relative account of a contact	<p>--1) Open your web browser, enter the “web” interface. (For details, you can refer to 7.Web Settings.)</p> <p>--2) Open “Contact” > “Phone book”, select the contact who are needed to be modified, click </p> <p>--3) Select the account in the drop-down column of the account, click “Submit” to complete it.</p>

Using Call Logs

Your phone maintains records of your missed, placed, and received calls.

If you want to...	Then...
View your call logs	<p>--1) Press MENU button > “Call history” > “Missed Calls”, “Received Calls”, or “Dialed numbers”</p> <p>--2) Use the navigation keys to view the call record information.</p>
Dial from a call log	Please refer to the previous part 4.Basic call handing – Placing a call.
Erase your call logs	<p>--1) If you want to delete a call record, you have to select this record from the logs and press Del;</p> <p>--2) If you want to delete an entire call record list, you have to select this record list from the logs and press DelALL</p>

Tips

- Each call log store up to 20 entries on SayHi ES330 IP phone.

6. Keypad Instruction

SayHi series IP phones can be configured in two ways. The first you can use the phone keypad where you can settings for you IP phones, the other you can log in to User Options web pages where you can settings for you IP phones.

Use phone keypad to setting. Press **MENU** or **OK** button to the main menu, Use the navigation keys to select menu, press **OK** button to confirm menu selections, press **C** button or **Del** to delete input information.

Language

SayHi ES330 IP Phone supports Simplified Chinese and English.

If you want to...	Then...
To change the language via Phone interface	--1) Choose "System setting" > "Phone setting" > "Language"; --2) Scroll through the list of available languages. --3) Press OK button or Modif when the desired language is highlighted. The language appears on the graphic display will be changed to the one you chose.

SIP Account Settings

SayHi ES330 series IP phone make calls based on sip accounts, SayHi ES330 series IP phones can support 2 independent SIP account, each account can be configured to different SIP server.

If you want to...	Then...
-------------------	---------

Create an SIP account	<p>--1) Choose “System setting” > “Advanced setting” ;</p> <p>--2) Enter the password required (The default is empty) ;</p> <p>--3) Choose “SIP” > “Account sip” ;</p> <p>--4) Choose one of the account you want to setting, you can configure the following parameters</p> <ul style="list-style-type: none"> -Enable account*: choose Enable -Display Name: The name displayed on the screen -User Name*: the account matched with the SIP server. (extension number) , -Authen usr: the Authenticated users matched with the SIP server. (The default With the same account) -user pwd*: the user password matched with the SIP server -Description: description of this account, -SIP1*: the primary SIP server, By default all calls through the server, -SIP2: the secondary SIP , When the primary server is unavailable ,use the SIP server -Refresh time: Registration refresh interval, the minimum value is 20 The default value is 3600. <p>--5) Set up the above parameters, Press Submit softkey to saves settings, Complete the account creation ;</p> <p>* Note: the parameters with the * mark must be set.</p>
Disable sip account	<p>--1) Choose “System setting” > “Advanced setting” ;</p> <p>--2) Enter the password required (The default is empty) ;</p> <p>--3) Choose “SIP” > “Account sip” ;</p> <p>--4) Choose “Enable account” > “Disable” ;</p> <p>--5) Press Submit soft key</p>

Network Setting

If you want to ...	Then...
network setting	<p>--1) Choose “System setting” > “Advanced setting” ;</p> <p>--2) Enter the password required (The default is empty) ;</p> <p>--3) Choose “Network”, you can configure the following parameters:</p> <p>-Type: static IP or DHCP</p> <p>-IP: enter IP address , Note: Do not duplicate the IP address with other devices on the network</p> <p>-Mask: enter appropriate subnet mask</p> <p>-GW: enter appropriate gateway</p> <p>- DNS1: enter IP address of the primary DNS server</p> <p>- DNS2: enter IP address of the secondary DNS server</p> <p>-Web port: the default Web port is 80,if you change it(for example change it to 88),you must use IP and Web port to login the web page (for example http://192.168.0.200:88).It will take effect on next reboot.</p> <p>-Telnet port: the default Telnet port is 23,if you change it(for example change it to 2003),you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.</p>

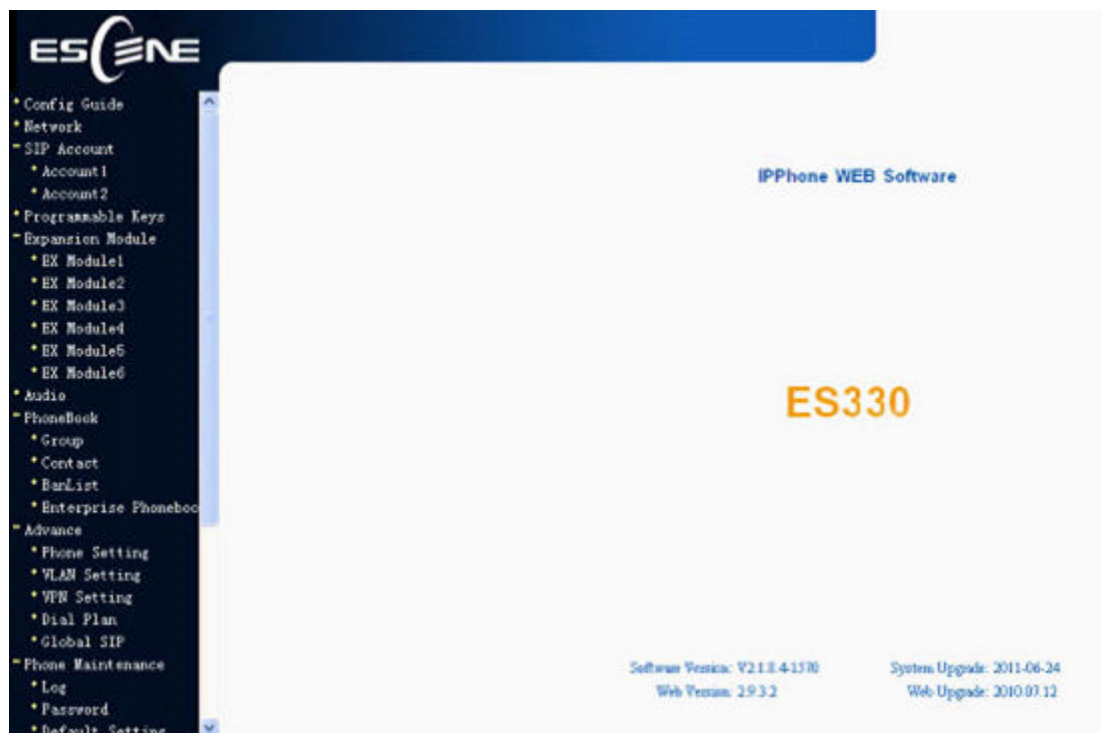
Customizing Rings and Volume

If you want to...	Then...
Change the ring tone	<p>--1) Choose “System setting” > “Phone setting” > “Ring type” ;</p> <p>--2) Press navigation to choose ring tone ;</p> <p>--3) Press Play softkey Choose a ring tone to play a sample of it.</p> <p>Press Stop softkey to Stop Playing</p> <p>Press OK or Selec softkey to set the ring tone,</p> <p>Press Back softkey to cancel</p>

Adjust the volume level	<p>--1) Choose “System setting” > “Phone setting” > “Volume setting”;</p> <p>--2) You can adjust the volume level of following types</p> <ul style="list-style-type: none">-Ring volume: Phone call ring volume,-Handset volume: Handle output volume,-Handset mic volume: Handle input volume,-Speaker volume: Hands-free speaker output volume,-Speaker mic volume: Hands-free input volume,-Headset volume: Headphone output volume,-Headset mic volume: Headset microphone input volume
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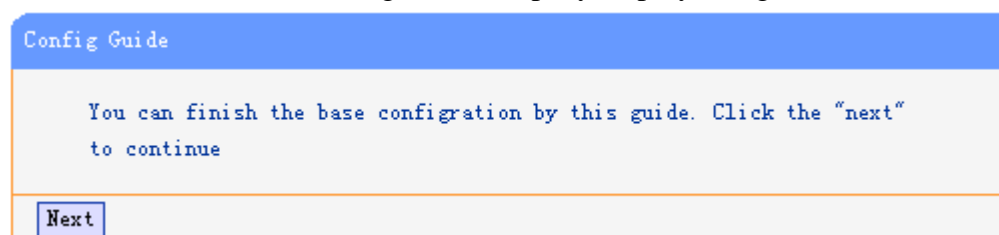
7.Web Settings

We can configure IP Phone more handy through web setting. Press OK button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example <http://192.168.0.200>) into the address bar of web browser. The default bgin name and password are both “root”.



Config Guide

You can finish the base configuration step-by-step by this guide.



When press 'next',you can configure the Network parameters for the phone,You can chose other option,such as static IP or PPPoE.

Network

IP Type
☒ DHCP
☐ Static IP
IP Address: 192.168.0.200
Netmask: 255.255.255.0
Gateway: 192.168.0.1
☐ PPPoE
Username: kk
Password: ●●●●●●
MTU: 1500 Default: 1500

DNS
☒ Automatic Get DNS
☐ Manual DNS
Primary DNS: 192.168.0.10
Secondary DNS: 0.0.0.0

MAC Address
MAC Address: 00:26:8b:00:51:69

Port Management
HTTP Port: 80
Telnet Port: 23

OutboundProxy Server
: ☒ off ☐ on
Server IP: *
.....

After config the network parameter,press next,then you can config sip account for the phone.

Account

SIP

Username: *

Password: *

SIP Server:

Attention: If you want to get more configuration information, please click to the appropriate Web page.

Press Finish, the base configuration of the phone is complete, now you can use the phone to call with sip.

Network

You can config the network parameters for the phone on the web page.

Network

IP Type

☐ DHCP

☒ Static IP

IP Address:

Netmask:

Gateway:

☐ PPPoE

Username:

Password:

MTU: Default: 1500

DNS

☐ Automatic Get DNS

☒ Manual DNS

Primary DNS:

Secondary DNS:

MAC Address

MAC Address:

Port Management

HTTP Port:

Telnet Port:

OutboundProxy Server

Choose network, you will find the following parameters:

Field	Description
DHCP	Config the phone get ip info from DHCP server
IP Address	Config the ip manual for phone
Netmask	Config the netmask manual for phone
Gateway	Config the gateway manual for phone
Username (pppoe)	The pppoe username
Password (pppoe)	The pppoe password
MTU (pppoe)	The mtu for pppoe,default is 1500
Primary DNS	The primary DNS server
Secondary	The secondary DNS server
MAC Address	Display the MAC of the phone
HTTP Port	The default web port is 80,if you change it(for example change it to88), You must use IP and Web port to login the web page(for example http://192.168.0.200:88). It will take effect on next reboot.
Telnet Port	the default Telnet port is 23,if you change it(for example change it to 2003),you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.

SIP Account

The phone attempts to register to the SIP server using the account/registrar data provided by the automatic or manual initialization

Account1

SIP

Enable: ☒

Account Mode: VOIP ▼

Display Name: 2209

Username: 2209 *

Authenticate Name: 2209

Password: •••• *

Label: 2209

SIP Server: 192.168.3.101

Secondary server:

OutboundProxy Server:

NAT Traversal: Disable ▼

STUN Server:

: ☒ SIP ☐ TEL

Subscribe Period: 3600 Default: 3600s, Min: 20s

Register Expire Time: 3600 Default: 3600s, Min: 40s

SIP Transport: ☒ UDP ☐ TCP ☐ TLS

Call

Amount Of Line Account Used: 2 (Default: 2)

Do Not Disturb: ☒ off ☐ on

Choose one Account, you will find the following parameters:

Field	Description
Enable	You can choose on/off to enable/disable the line.
Account Mode	You can choose VOIP/PSTN,when you want to use as PSTN,you should select PSTN and plug the RJ-11 phone line into the SOR port of the phone.
Display Name	It is showed as Caller ID when making a phone call
Username	It is a username provide by SIP Server
Authenticate Name	It is authenticated ID for authentication
Password	It is a password provide by SIP Server
SIP Server	Server for registration, provided by administrator
Register Expire Time	IP phone automatically registered every time
Amount Of Line Account Used	The line key of account used,default is 2

Choose one Account, you will find the following parameters:

Programmable Keys

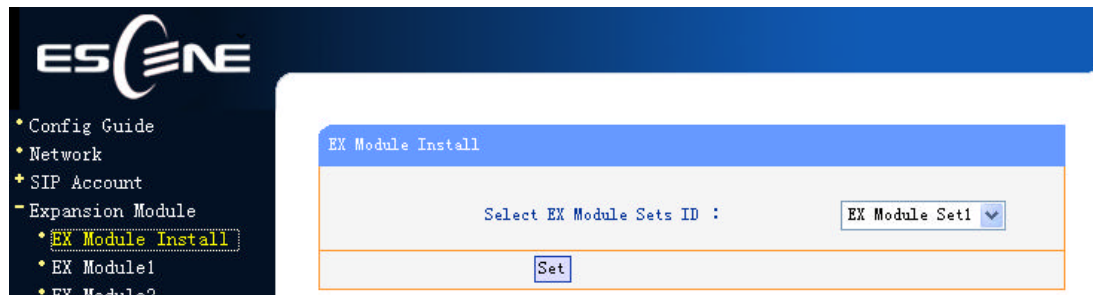
Choose Programmable Keys, you will find the following parameters:

Field	Description
Speed Dial(Mode)	Use specific Key as Speed Dial key
Asterisk BLF(Mode)	Use specific Key as BLF key
Speed Dial Prefix(Mode)	Use specific Key as Speed Dial Prefix key
DTMF	Use specific Key as DTMF key
Asterisk BLA	Use specific Key as BLA key

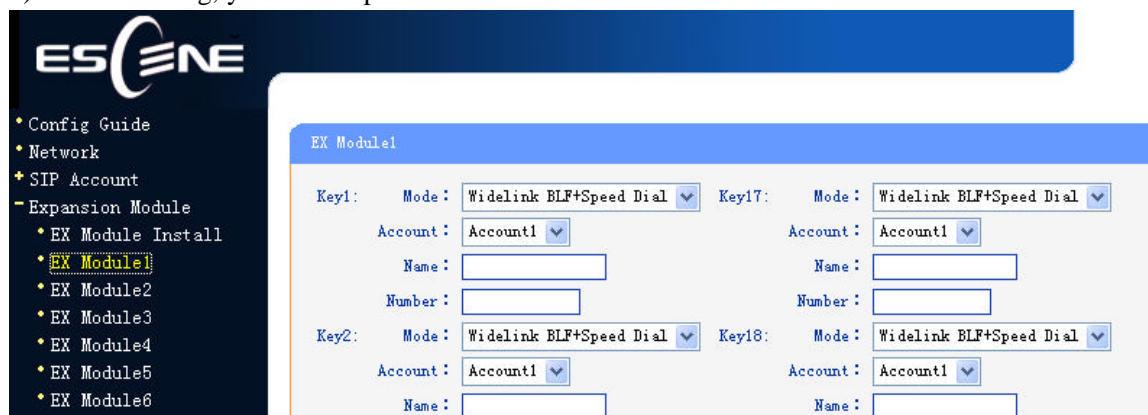
Expansion Module

Expansion module is extended Hotline function; you can believe it support more hotline by using Expansion module. You can refer to *2.Connecting Your Phone* about connecting of Expansion module.

After connecting Expansion module to a phone, you can install it as follow:



- 1) Click “Expansion Module”, select a module that you install (the one you connect to IP phone directly is Module1, the one connect to Module1 is Module2, and so on) and click “Set”.
- 2) After installing, you can set parameters of each Extension as follow:



Expansion module	
Key n	Each Expansion module supports 32 keys.
Mode	Five modes: --Speed Dial: Enable speed dialing in this key; --Asterisk BLF: Enable BLF in this key; --Speed Dial Prefix --DTMF --Asterisk BLA
Account	A SIP account relates to this key, another word, you will call this hotline by this SIP account.
Name	Description of this hotline.
Number	Number relates to this key.

*Regarding the settings of Expansion module, please confirm the model of your phone is DS3xx_E/DS3xx_PE.

Audio

The IP phone supports the following voice codecs: G.722, G.711A, G.711U, G.723, and G.729A.

You can enable/disable the desired codecs via Web interface. Please contact your system administrator for more details about the codecs.

To enable/disable the codecs:

- 1) Choose Audio-> Audio Codecs

Audio

Tone

Dial Tone: Ring Volume (0~9):

Output Volume (1~9)

Handset Volume: Handset Mic Volume:

SpeakerPhone Volume: SpeakerPhone Mic Volume:

Headset volume: Headset Mic Volume:

Voice Codec

Payload Length: ms High Rate of G723.1: ☐

Other

VAD: ☐ Echo Suppression Mode: ☐

Ring

Ring Type:

Uploading Ring Tone

((Please upload a ring tone with G711 audio coding, and the size must less than 300k.))

Audio Codecs:

enableCode

- 2) Use the navigation keys to highlight the desired one in the Enabled/Disable Codecs list, and press

the / to move to the other list.

- 3) Choose Submit to save the change.

Of course, you can control the voice bulk in this choose.

PhoneBook

Group

You can add, edit and delete group in a phone book on web page of ES330.

- 1) Click “PhoneBook” > “Group”,

Group


Click the groupname you can modify or delete the member of the group

ID	Operation	Group Name	Group Member	Description
Attention: If you Click 'Delete Group' or 'Delete All Group', the member of group can not within a group, please click the group and delete the group.				

Add Group Delete All Group

If you want to add a Group, you just ought to click ‘Add Group’.

You can edit an existed Group by click .

You can delete an existed Group by click , if you want to delete all Groups, you just ought to click 'Delete All Group'.

- 2) When you add a group or edit an existed group, you can set several parameters as follow:

Group

ID: 1 Description:

Group Name:

Submit Cancel

Group	
ID	Serial number of a group
Description	Description of a group
Group Name	Name of a group

Contact

You can add, edit and delete contact in a phone book on web page of ES330.

The phonebook can storage 300 contact entry.


- 1) Click “PhoneBook” > “Contact”,

Contact

ID	Operation	Name	Phone	Group
Add Contact Delete All Contact				

If you want to add a Group, you just ought to click ‘Add Contact’.

You can edit an existed Contact by click .

You can delete an existed Contact by click , if you want to delete all Contacts, you just ought to click 'Delete All Contact'.

- 2) When you add a Contact or edit an existed Contact, you can set several parameters as follow:

Contact	
Serial Number	Serial number of a contact

First Name	The First Name of a contact
Last Name	The Last Name of a contact
Mobile Number	The Number1 phone number of a contact
Office Number	The Number2 phone number of a contact
OtherNumber	The Number3 phone number of a contact
Group	You can assign a contact to a specific group. If there isn't any group set on the phone, the group is None by default.
Account	Select a SIP account relating this contact, that is you can dial to the contact from this SIP account.

BanList


You can add, edit and delete banlist in a phone book on web page of ES330.

- 1) Click "PhoneBook" > "BanList",

BanList					
ID	Operation	Name	Phone	Description	Account
<div> <div>Add BanList</div> <div>Delete All BanList</div> </div>					

If you want to add a BanList, you just ought to click 'Add BanList'.

You can edit an existed BanList by click .

You can delete an existed BanList by click , if you want to delete all BanLists, you just ought to click 'Delete All BanList'.

- 2) When you add a BanList or edit an existed BanList, you can set several parameters as follow:

BanList	
Serial Number	Serial number of a BanList
Description	Description of a BanList
First Name	The First Name of a ban contact
Last Name	The Last Name of a ban contact
Mobile Number	The number1 phone number of a ban contact
Home Number	The number2 phone number of a ban contact
Office Number	The number3 phone number of a ban contact
Account	Select a SIP account relating this ban contact, that is the ban contact can't dial to this SIP account.

Enterprise Phonebook

You can download Enterprise Phonebook from this web interface. But you should do second develop on the sip server to enable this function completely.

If the sip server no add some function to hold this option ,this option can be used.

Enterprise Phonebook

☐ Auto Download Enterprise Phonebook

Server IP:

Password:

Submit

Advance

Phone Setting

You can use phone setting to set the time,qos,port Mirroring for the phone.

Phone Setting

Basic

Called No AnswerTime: ☒ 30 s ☐

DTMF: ☒ RFC 2833 ☐ Inband ☐ SIP Info ☐ Auto

: ☒ # ☐ %23

RFC 2833 PayLoad:

BackLight: ☐ off ☒ Time s (Default :60, Min:1, Max:255)

PSTN Setting

PSTN Ring Type: ☐ PSTN Ring ☒ VOIP Ring

PSTN Prefix Code:

VOIP Prefix Code:

Hook: ☐ off ☒ on

Hook Frequency: (Default:500 min:100 max:1600)

Call

Hot Line Function: ☒ off ☐ Immediately Hot Line ☐ Delay

Hot Number:

Auto Answer: ☒ off ☐ on

Pickup Code:

Message:

Booking Voicemail: ▼

Hang voice Play: ☐ off ☒ on

VOIP Call Forward

Always: ☒ off ☐ on Number:

When used Phone Setting option,you can set several parameters as follow:

Phone Setting	
DTMF	The DTMF transmitted mode,include RFC 2833,Inband,SIP Info
BackLight	The backlight of the phone LCD
Set Time Mode	The mode of set time for phone,include SNTP/SIP Server/PSTN/Manual
Daylight Saving Time	Enable/disable the DST for the phone
Time Format	You can use 24 hour time format or 12 hour time format
Time Zone-GMT	You can select different time zone for the phone
Manual Setting	This used to manual set time for the phone
QoS	The qos priority,support diff-serv and precedence
Network Packet Mirroring	When select on,then you can capture the phone's packet use notebook which connect to pc port of the phone

VLAN Setting

You can add the phone and PC to different VLAN used VLAN Setting option.

VLAN Setting	
Voice Enable VLAN: <input type="checkbox"/> VID: <input type="text" value="0"/> (0~4094) Priority: <input type="text" value="0"/> (0~7)	PC Enable VLAN: <input type="checkbox"/> VID: <input type="text" value="0"/> (0~4094) Priority: <input type="text" value="0"/> (0~7)
<input type="button" value="Submit"/>	

When used VLAN Setting option, you can set several parameters as follow:

VLAN Setting	
Enable VLAN	You can enable/disable vlan for phone and pc
VID	The vlan you want the phone or pc to join

VPN Setting

VPN Setting	
Enable VPN: <input type="checkbox"/> VPN Type: <input type="text" value="L2TP"/>	
L2TP	VPN Server Addr: <input type="text"/> VPN User Name: <input type="text"/> VPN Password: <input type="text"/>
<input type="button" value="Submit"/>	

IF you need to setup a VPN Setting, you should fill below options.

When used VPN Setting option, you can set several parameters as follow:

VLAN Setting

Enable VPN	You can enable/disable VPN for phone and pc
VPN Type:	There is one choose you can choice.
VPN Server Addr	VPN server'ip
VPN User Name	VPN iser's name
VPN User Name	A password be used foe authentication

Dial Plan setting



If you want to setup a dial plan, you can click "Dial Plan".

Dial Plan

☒ Send Key: ☐ * ☒ #

☐ Dial Length:

☐ No Dial Timeout:

ID	Operation	Prefix	IP Address	Description
1	 	209	192.168.2.83	

Click "add rule" to entry this interface.

Dail Rule And Routing			
ID:	1	Description:	
IP:		Port (Default 5060):	5060
Prefix:			
Called Insert Number:	Disable	Called Delete Number:	Disable
Position:		Position:	
Number:		Length:	
Caller Insert Number:	Disable	Caller Delete Number:	Disable
Position:		Position:	
Number:		Length:	
<p>(Note: When you want to add code and delete at the same time, you can add code first, after that base on the number you add, decide the position and length of the delete code.)</p>			
<input type="button" value="Submit"/> <input type="button" value="Cancel"/>			

Dial Plan Setting	
ID	Dial Plan ID
IP	The ip of a phone which you want to call
prefix	The number which you need to press actually if you want to call the phone
Called Insert Number	There have two option, Enable or Disable.
Position	Which position you want insert the number
Number	Waht number you want to insert
Called Delete Number	There have two option, Enable or Disable.

Tips

(Note: When you want to add code and delete at the same time, you can add code first, after that base on the number you add, decide the position and length of the delete code.)

Global SIP

You also can setup the SIP server on Global SIP.

Global SIP

SIP
SIP Server:

Secondary server:

Proxy Server
OutboundProxy Server:

STUN
STUN Server:

Others
Register Expire Time: s Default: 3600s, Min: 40s
Local SIP port: (Default: 5060)
SIP Transport: ☒ UDP ☐ TCP ☐ TLS
RTP Port Range: --
SUB Expire Time:

Submit

PasswordPhone Maintenance

Log

If you need to catch a debugging Level, you need setup on this interface.

Log

☐ No Record
☐ Call:
☒ SIP
☐ DSP
☐ LCD

Submit

You can change the password used to login phone GUI in Password option.

In Password option, you can set several parameters as follow:

Password	
Username	The login username of the web page
Old Password	The old password used to login of the web page
New Password	The new password used to login of the web page
Confirm Password	The new password used to login of the web page
Administrator	Login phone web page used administrator privileged
User	Login phone web page used general user privileged

Default Setting

You can load the phone to the factory default setting in default setting option.

Press the 'Reset to Factory Setting' option, the phone will load to factory default setting on next reboot.

Auto Provision

when you open the auto provision function, the phone will auto provision if the phone detect a higher software or kernel which are put on the software server. the detail information about auto provision you can see the appendix.

Auto Provision

Auto Provision: ☒ on ☐ off

☒ DHCP Option

Option: (Default :66, Min:1, Max:254)

Protocol:

Software Server URL:

Username:

Password:

☒ Auto Download Software

☒ Auto Download Kernel

☒ Auto Download Config File

☐ Broadsoft Compatiblity

☒ Auto Download Expansion

☒ Auto Download Enterprise Phonebook

☒ Auto Download Personal Phonebook

☒ Booting Checked

Auto Provision Frequency: Hour (Default :7 days, Max:30 days)

Auto Provision Time:

AES Enable: ☒ off ☐ on

AES Key:

When use auto provision, you can set several parameters as follow:

Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	The protocol use for auto provision,it include tftp/http/ftp
Software Server URL	The server address of the auto provision
Username	The username provide by provision server
Password	The password provide by provision server
Auto Download Software	This used to auto download software from server
Auto Download Kernel	This used to auto download kernel from server
Auto Download Config File	This used to auto download config file from server
Broadsoft Compatiblity	This used to compatible the broadsoft format's config file

Auto Download Expansion	This used to auto download expansion's config from server
Auto Download Enterprise Phonebook	This used to auto download enterprise phone from server
Auto Download Personal Phonebook	This used to auto download personal phonebook from server
Bootling Checked	This used to checked the auto provision when phone booting
Auto Provision Frequency	This used to set the time interval for auto provision
Auto Provision Time	This used to the specific time for auto provision
AES Enable	You can enable/disable AES encrypt for auto provision
AES Key	The key of the AES
Auto Provision Now	This used to do auto provision immediately

FTP Upgrade

You can upgrade the software,kernel and configure file for the phone use ftp.

FTP Upgrade (Attention: Do not cut off the electricity when Upgrade!!)

Server IP:
Filename:
Username:
Password:
Software Upgrade:
Kernel Upgrade:

Note: It's no necessary to input filename when backup.

Configuration:
Phone Book:
EXT Module:

When use ftp upgrade, you can set several parameters as follow:

FTP Upgrade	
Server IP	The ip address of the ftp server
Filename	The name of the file want to download from ftp server

Username	The username provide by ftp server
Password	The password provide by ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

TFTP Upgrade

You can upgrade the software,kernel and configure file for the phone use tftp.

TFTP Upgrade (Attention: Do not cut off the electricity when Upgrade!!)

Server IP:

Filename:

Software Upgrade:

Kernel Upgrade:

Note: It's no necessary to input filename when backup.

Configuration:

Phone Book:

EXT Module:

When use tftp upgrade, you can set several parameters as follow:

TFTP Upgrade	
Server IP	The ip address of the tftp server
Filename	The name of the file want to download from ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

HTTP Upgrade

You can upgrade the software, kernel and configure file for the phone use http.

HTTP Upgrade (Attention: Do not cut off the electricity when Upgrade!!)

HTTP Upgrade:

Select a File:

Software Upgrade:

Kernel Upgrade:

Configuration:

PhoneBook:

EXT Module:

Log:

All Config File:

When use http upgrade, you can set several parameters as follow:

HTTP Upgrade	
Select a File	Browse the software/kernel/config file you want to upgrade from http
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used upload/download to upload/download the configure file of the phone
Phone Book	You can used upload/download to upload/download the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

Reboot

You can use reboot option to reboot the phone.

Reboot

Attention: When click this button this equipment will be reboot, web service will be interred, please connect again.

When you press 'Reboot', the phone will reboot.

Phone Status

You can see the currently status of the phone when use Phone Status option.

Phone Status

System Run Time 0 Day2 Hour12 Minute30 Second

Register status

Account1: Registered

Account2: Unregister

EX Module1: Off Line

EX Module2: Off Line

EX Module3: Off Line

EX Module4: Off Line

EX Module5: Off Line

EX Module6: Off Line

Network Status

Connection: Dynamic

IP Address: 192.168.2.147

Netmask: 255.255.0.0

Gateway: 192.168.0.10

Primary DNS: 192.168.0.10

Secondary DNS: 0.0.0.0

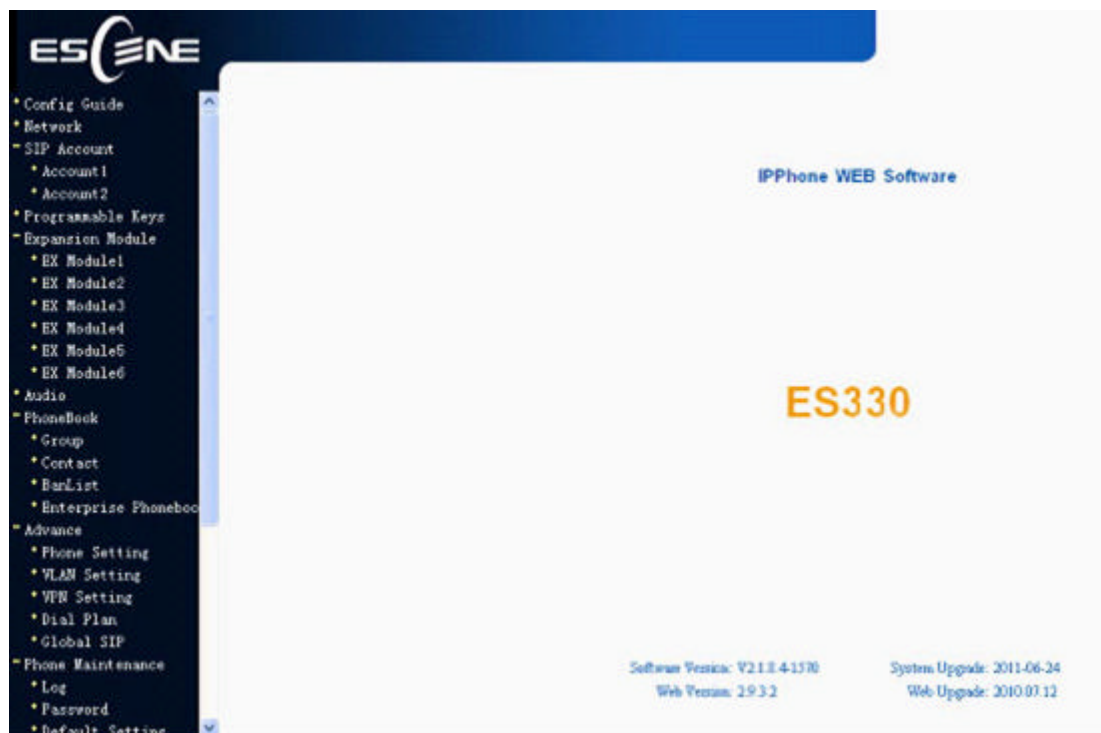
VPN IP Address:

Hardware

Hardware ID: 11

About

You can see the phone model when used About option.



Attention:

On this interface ,you can see the software and kernel which we used for test and this user_manual is written base on this software and kernel.

This software version is V2.1.8.4-1570

This kernel version is v2.2.3

Appendix :

Auto Provision

Pre-configuration on TFTP/HTTP/HTTPS/FTP Server

When the software or kernel auto-provision is enabled and want to run, IP Phone will check the software and kernel version at first, so we need make some pre-configuration on the provisioning server.

Auto Provision for Software:

1. Create a notepad file named "**F000X00.cfg**" (the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "F000600.cfg");

**Named rule of the file:*

F00600.cfg: for ES620, ES610 and DS622;

F00400.cfg: for ES410 and DS412;

F00300.cfg: for ES310 and DS312;

F00200.cfg: for ES210 and DS212.

2. Open the notepad file "F000X00.cfg" and write the new software name in it, for example,

S_ES6xx_version2.0.4.6: for ES620, ES610 and DS622;

S_ES410_version2.0.4.6: for ES410 and DS412;

S_ES310_version2.0.4.6: for ES310 and DS312;

S_ES210_version2.0.4.6: for ES210 and DS212

Write down the new version you want to upgrade and save it on your provisioning server.

**Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your software would be not available.*

3. After it, upload the new software to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Auto Provision for Kernel:

1. Create a notepad file named "**K000X00.cfg**" (the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "K000600.cfg");

**Named rule of the file:*

K00600.cfg: for ES620, ES610 and DS622;

K00400.cfg: for ES410 and DS412;

K00300.cfg: for ES310 and DS312;

K00200.cfg: for ES210 and DS212.

2. Open the notepad file "K000X00.cfg" and write the new kernel name in it, for example,

K_uImage_600.bin_version2.1.6: for ES620, ES610 and DS622;

K_uImage_400.bin_version2.1.6: for ES410 and DS412;

K_uImage_300.bin_version2.1.6: for ES310 and DS312;

K_uImage_200.bin_version2.1.6: for ES210 and DS212

Write the new version you want to upgrade and save it on your provisioning server.

**Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your kernel would be not available.*

3. After it, upload the new kernel to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Configuration files on TFTP/HTTP/HTTPS/FTP Server

- **Name of configuration file:**

The configuration file on the provisioning server is named as the MAC address of IP phone itself. Escene's IP phones support two different configuration files for auto-provision:

1. Normal Configuration file:

Normal Configuration file is the configuration file of your Escene IP phone. You can download it from your phone (You can see the following chapter to see how to download a configuration file from Escene IP phone) and modify by yourself. If the IP phone's MAC address is 00:11:22:33:44:55, the normal configuration file of it should be **001122334455.xml**.

2. Broadsoft Configuration files:

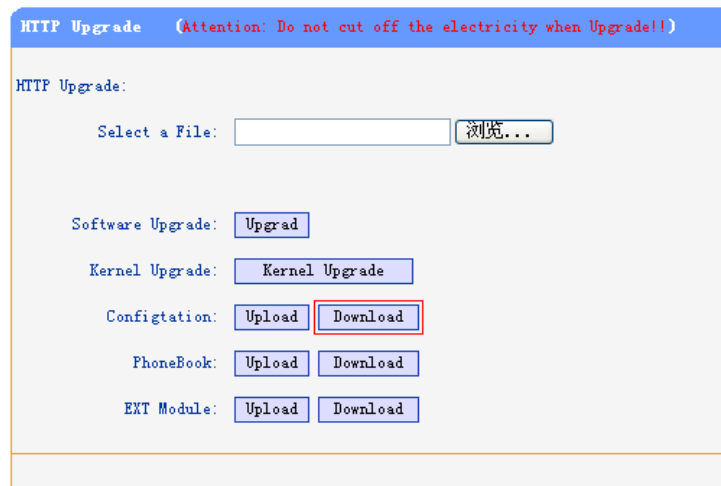
Broadsoft Configuration files support the format of Broadsoft IP-PBX. However, you can use them for provisioning. There are two files should be set on your provisioning server, they are also named by the MAC address of your phone

- 1) **001122334455.cfg**: a configuration file for system settings, for example, network, audio and so on.
- 2) **001122334455.txt**: a configuration file for SIP accounts.

- **Download a configuration file from your phone:**

You can download a configuration file from your phone by HTTP as follow:

1. Open the web page of your IP phone, click "Phone Maintenance">"HTTP Upgrade";
2. Then click "Download" of Configuration:



3. If you want to use this file to auto-provision, you just need to modify it by yourself and rename it to the MAC address of your IP Phone with .xml suffix.

Extern.xml file on TFTP/HTTP/HTTPS/FTP Server

The Extern.xml includes the settings of programmable buttons on the phone and all Expansion Modules. All the phones can download the settings from a same file and they will have the same settings (for example, Speed-dial, BLF and so on).

**You can't rename the file on the provisioning server. The file name is fixed to Account1_Extern.xml.(Account1 is the first account you register)*

Phonebook on TFTP/HTTP/HTTPS/FTP Server

Escene IP phone supports Enterprise Phonebook and Personal Phonebook.

- **Enterprise Phonebook:**

Enterprise Phonebook is used for all staffs in your office. All phones will download a common phonebook for all staffs. The file's name must be

Enterprise_Phonebook.xml on your provisioning server and you can not rename it.

- **Personal Phonebook:**

Personal Phonebook is individual for each IP phone. The file on your provisioning server is named by the first account of your IP phone. If the IP phone's first account is 1287, the Personal Phonebook of this phone is **1287_Phonebook.xml**.

Automatic Provisioning using DHCP Option 66

The following steps will descript auto-provision by TFTP. You also can use HTTP and FTP for auto-provision with our phones.

DHCP Server: (Microsoft Windows 2003 server)

1. Start up the “DHCP Management Console”;
2. Expand the DHCP scope which will contain the phones
3. Right-click on the “Scope Options” node
4. Select “Configure Options”
5. In the “General” tab, scroll down the list of options and identify the option labeled “066 Boot Server Host Name”
6. Enable the “066 Boot Server Host Name” and enter the string value according to the examples discussed previously

string value:
192.168.0.201(TFTP Server IP Address);

7. Click the “OK” button

IP Phone:

1. Input the IP Phone’s IP Address in browser;
2. Enter user and password with “root” then open the web page;
3. Click “Phone Maintenance” and select “Auto Provision”;
4. Select like as follows:

5. Click “Submit” to save it.

Auto-Provision via fixable TFTP/HTTP/HTTPS/FTP Server

IP Phone:

1. Input the IP Phone’s IP Address in browser;
2. Enter user and password with “root” then open the web page;
3. Click “Phone Maintenance” and select “Auto Provision”;
4. select like as follows:

Auto Provision

Auto Provision: ☒ on ☐ off

☒ DHCP Option

Option: (Default :66, Min:1, Max:254)

Protocol:

Software Server URL:

Username:

Password:

☐ Auto Download Software

☒ Auto Download Kernel

☒ Auto Download Config File

☐ Broadsoft Compatibility

☒ Auto Download Expension

☒ Auto Download Enterprise Phonebook

☒ Auto Download Personal Phonebook

☒ Booting Checked

Auto Provision Frequency: Hour (Default :7 days, Max:30 days)

Auto Provision Time:

AES Enable: ☒ off ☐ on

AES Key:

It supports three protocols in Auto-Provision:TFTP,HTTP and FTP.

The format with provisioning server URL is:

TFTP:

TFTP://192.168.0.201(192.168.0.201 is the default Server IP address)

HTTP:

HTTP://192.168.0.201

HTTPS:

HTTPS://192.168.0.201

FTP:

FTP://192.168.0.201

Username: the user to login FTP/HTTP/HTTPS server

Password: the password of the user using to login FTP/HTTP/HTTPS server

**Username and password are available in FTP/HTTP/HTTPS only (unavailable in TFTP).*

Auto Download Software:

Download software from server and upgrade it automatically.

Auto Download Kernel:

Download kernel from server and upgrade it automatically.

Auto Download Config File:

Download configuration file from server and update it automatically.

Broadsoft Compatibility:

If you select this function, you need to put two configuration files (with Broadsoft format) on the provisioning server. Otherwise, you can download the configuration file from your phone via HTTP (regarding the steps, you can refer to “*Download a configuration file from your phone*” in this document.), modify it and upload it to the server for auto-provision.

Auto Download Expansion:

Download configuration file of the Programmable buttons on your phone or Expansion Modules automatically.

Auto Download Enterprise Phonebook:

Download Enterprise Phonebook from server and update it automatically.

Auto Download Personal Phonebook:

Download Personal Phonebook from server and update it automatically.

Bootling Checked:

Check all items you had selected and upgrade/update them when the phone boot

Auto Provision Frequency:

The auto provision Frequency which you want.

Auto Provision Time:

The time you want to execute auto-provision.

Examples of Auto Provision Frequency and Time

- 1) When you set the **Auto Provision Frequency** and disable **Auto Provision Time** (set to None), the Auto Provision function will work after the **AutoProvision Frequency**;

- 2) When you set both **Auto Provision Frequency** and **Auto Provision Time**, for example:

You set the **AutoProvision Frequency** to 24 hours, and the **Auto Provision Time** to 2:00 at 8:00 today (1, Jan), it will pass 24 hours at first and work at the nearest 2:00, it means that the Auto Provision function will work at 2:00 on the day after tomorrow (3, Jan).

Therefore, if you want this function work at 23:00 tonight and it is 8:00 now, you need to set the **Auto Provision Frequency** to 0 hours and the **Auto Provision Time** to 23:00.

AES Encryption:

AES encryption is used for all the setting files of your phone (include configuration file, Expansion file, Enterprise/Personal Phonebook etc. You just need to enable the

AES Encryption function and input the AES Key matching the one on your server on.



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